Performance Improvement of AMBE 3600 bps Vocoder with Improved FEC

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***Abstract*—Efficiency and performance of the heavily used electronic devices in the field are always open for debate. As the technology advances, efficiency and performance of the electronic devices increase. Digital communication systems are also getting their share of this development trend. Digital communication systems, such as Digital Private Mobile Radio, Digital Mobile Radio/MotoTRBO, Association of Public-Safety Communications Officials-International Project 25 and Icom-Kenwood NEXEDGE use half rate Advanced Multi-Band Excitation (AMBE) 3600 bps vocoder to provide clean and intelligible voice communication service. Although this half rate vocoder incorporates Forward Error Correction (FEC) coding to protect voice frames, its error correction performance does not meet today's standards. To improve the FEC performance of the vocoder, in this work, we assess the FEC portion of the vocoder and propose a better performing FEC scheme. The proposed 2/3 rate convolutional code with vocoder frame length reduction provides a 4.41 dB coding gain in the high signal-to-noise region compared to AMBE FEC while preserving audio quality. Since all works are conducted around the vocoder section, improvements can be easily implemented in existing digital communication systems and standards.**

Keywords- Digital mobile radio; forward error correction; perceptual evaluation of speech quality; punctured convolutional coding; speech codec; vocoder.

I. INTRODUCTION

From small sites to huge organizations, mobile communication systems have evolved from analog to digital in order to keep up with the fast pace of the modern world. Since digital radios provide better sound quality and extensive data services compared to their analog counterparts, they are commonly used in areas where people need wireless communication. There is a large number of digital radio systems, which are already available, such as Digital Private Mobile Radio (dPMR) [1], Digital Mobile Radio (DMR) [2]-[3], NXDN [4], APCO P25 [5], digital smart technologies for amateurs (D-Star) [6], etc. All digital radios incorporate at least one type of vocoder to provide voice services over digital communication channel. There is a vast number of speech coders currently available. Advanced Multi-Band Excitation Speech Codec (AMBE) [1]-[2] is one of them and it is used in many communication systems. AMBE half rate 3600 bps vocoder is used in the following digital radio systems: dPMR, DMR/MotoTRBO, APCO P25 and NXDN [1]-[6]. AMBE 3600 bps vocoder consists of 1150 bps Forward Error Correction (FEC) and 2450 bps vocoder data.

AMBE speech codec is a type of speech compression technique. It consists of encoder and decoder. It can encode samples of voice data to a compressed stream and can generate synthesized voice output bits from the compressed bit stream [7].

In theory, digital radios outperform their analog counterparts in terms of voice quality. We do not always have ideal conditions and also we do not always have high Signal-to-Noise Ratio (SNR) value. Due to attenuation and distortions in the communication channel, the overall Bit Error Rate (BER) performance of the digital radio degrades hence lowers the voice quality. Even though every 49 data bits of vocoder are accompanied by 23 FEC bits, the AMBE vocoder FEC cannot perform well. For this reason, limited communication range and bad audio quality can happen in noisy environments. To enhance the voice quality without modifying digital communication standards and protocols, the AMBE vocoder FEC should be improved. AMBE vocoder is an independent system which is realized in an isolated integrated circuit (IC) or a software library.

To improve the FEC performance of the AMBE vocoder, a better FEC can be employed. In [8], the authors show that replacing the AMBE forward error correction scheme with a combination of block code and 5/6 rate Punctured Convolutional Code (PCC) provides 3.35 dB additional gain in the high SNR region. But, we can further increase the coding gain using solely convolutional codes, hence increasing the audio quality and the communication range. Although Turbo codes provide more coding gain than convolutional codes, they are not suitable for short block size such as 49 bits [9] [10]. In addition, it is widely known that Low Density Parity Check codes work better when large block sizes are utilized [11]. Because we are dealing with a small block size, convolutional codes are selected in the proposed FEC scheme. In this work, we proposed and present an improved FEC which outperforms the AMBE standard and the FEC scheme proposed in [8].

In the following section, existing FEC schemes are described. In Section 3, the proposed FEC scheme is explained. In Section 4, the performance analysis is presented. In Section 5, the results of the paper and the conclusion are explained. In the last section, the future work is given.

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II. EXISTING FEC SCHEMES

AMBE 3600 bps vocoder is a very low-rate speech coder used for voice transmission. Due to the high compression ratio every bit of information in the compressed speech data stream has low or high importance, but not zero. Compressed audio is more vulnerable to bit errors compared to the sampled audio. Depending on which parameter bits are exposed to bit error, they either distort or impair the synthesized voice. So, vocoder frames must be protected very well to prevent audio loss in those digital radios using speech codecs like AMBE 3600 bps half rate speech coder.

FEC is a largely researched and advanced technique to detect and correct errors in data frames. Different error correction codes can recover different number of bits. If the data is received with a greater number of errors than that the employed FEC can recover, the decoder cannot reconstruct the received data correctly. In those conditions, catastrophic errors occur while reconstructing data frames. In an environment where BER is very low, the AMBE vocoder performs well and provides good voice quality at 3600 bps [7]. However, when the BER values are very high, the AMBE vocoder cannot correct the received vocoder frame errors, hence cannot reconstruct the audio. To make the AMBE vocoder more noise resistant, a better FEC scheme should be implemented.

One of the most commonly used vocoders, the half rate AMBE 3600 bps speech coder, is composed of 2450 bps voice and 1150 bps FEC data. By standard, AMBE coded voice is sent in 20 ms voice packets. Each voice frame is composed of 49 voice data bits and 23 FEC bits. The FEC used in the AMBE 3600 bps vocoder incorporates one extended Golay (24,12) code and one Golay (23,12) code. There is also data dependent scrambler and modulation key parameter involved in the FEC scheme. Standard FEC implementation is employed to protect the most sensitive 24 bits while the remaining bits are left unprotected [5]. Block diagrams of the AMBE 3600 bps FEC encoder and decoder are given in Figure 1 and Figure 2, respectively.

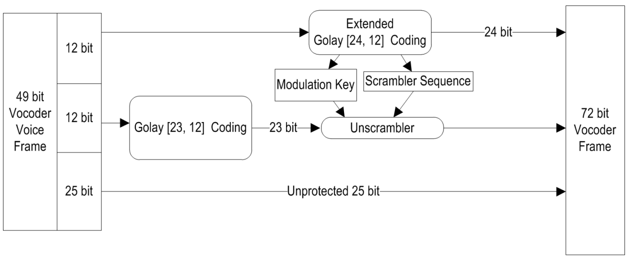

Figure 1. 3600 bps AMBE's vocoder FEC encoder scheme

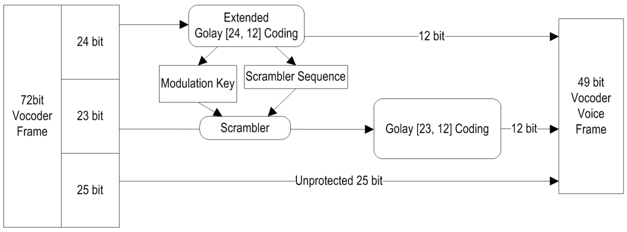

Figure 2. 3600 bps AMBE`s vocoder FEC decoder scheme.

As seen in Figure 1 and Figure 2, AMBE FEC scheme uses modulation key and data dependent scrambler derived from the first Golay code while encoding or decoding the second Golay code. If an irrecoverable error occurs in the first Golay code, the modulation key and the descrambler sequence cannot be calculated correctly. Wrong modulation key and descrambling sequence create more errors and escalade the overall erroneous bit count. Due to the high number of bit errors, irrecoverable frames cannot be decoded and discarded. Hence, the voice quality falls catastrophically and the received speech cannot be synthesized at all due to high BER. This chain reaction becomes self-inflicted destruction for the vocoder frames. In such situations, the AMBE vocoder offers comforting silence or frame repetition in the place of irrecoverable frames.

By standard, AMBE 3600 bps speech coder does not protect the whole voice frame from errors. Although less error sensitive or less significant vocoder parameter bits have less impact on synthesized voice quality, their effect is non-zero. All the bits in compressed speech have either a major or a minor effect on the overall synthesized voice quality. For a communication channel where the BER value is lower than P_b=10^{-5}, there is no noticeable change in voice quality and intelligibility. In contrast, when the BER value is higher than P_b=10^{-5}, the synthesized voice quality degrades and impairs intelligibility.

In order to improve the voice quality of the AMBE 3600 bps vocoder by making the vocoder more immune to errors, vocoder frames should have better protection than what AMBE provides. To enhance FEC performance, Golay (23,12) code along with 5/6 rate convolution code FEC scheme was implemented in the AMBE 3600 bps vocoder [8]. Hybrid FEC encoder and decoder block diagrams of the referenced work are given in Figure 3 and Figure 4, respectively.

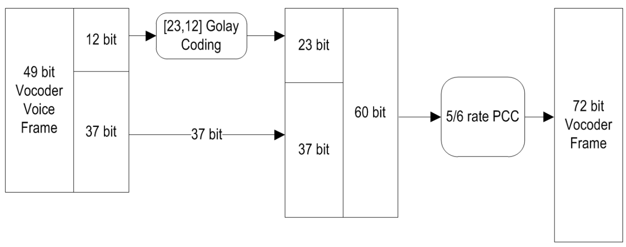

Figure 3. Hybrid FEC encoder block diagram for vocoder in [8].

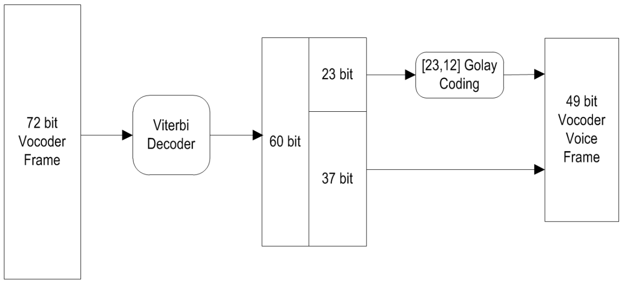

Figure 4. Hybrid FEC decoder block diagram for vocoder in [8].

In the given work, Golay (23,12) code adds additional 11 bits to 49 bits and makes 60 bits of data. After 5/6 rate convolutional code is applied, the frame becomes 72 bits. The referenced work incorporates block and convolutional codes in sequence to avoid making any modification to the AMBE vocoder frame while improving the FEC performance and synthesized audio quality.

III. PROPOSED FEC SCHEME

For further improvement on the AMBE FEC performance, instead of using solely block codes or using block and convolutional codes in sequence, utilizing purely convolutional code yields increased coding gain. To utilize a better convolutional code than in [8], the number of FEC bits in vocoder frame should be increased by 1 bit. As given above, the AMBE standard produces 23 FEC bits for every 49 vocoder data bits. In digital radio systems where vocoders and compressed data are used extensively, bit stealing is common practice to reduce data length to optimize the FEC or transmitted data rate [13] [14]. In order to steal 1 bit from vocoder voice frame data, more than 100 hours (over 19 million voice frames) of AMBE 3600 bps coded records have been analyzed in terms of individual bit probabilities in the vocoder voice frame. Bit probabilities in vocoder voice frame are given in Figure 5.

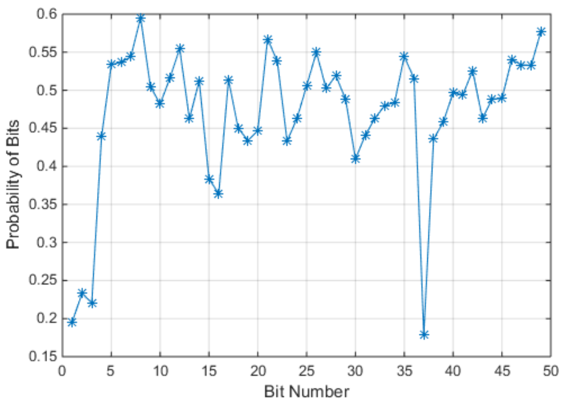

Figure 5. Bit probabilities in vocoder voice frame (calculated using 100 hours, over 19 million voice frames of AMBE 3600 bps record).

As seen in Figure 5, 37^{th} bit in voice frame is zero with a 82.104% probability. Additionally, 37^{th} bit is one of the less sensitive bits in AMBE 3600 bps voice frame and represents the least significant bit of 5-bit gain value [7]. Having said that, we decided to discard the 37^{th} bit in vocoder frames prior to encoding, and add 37^{th} bit back to the frame with zero value after decoding. After bit stealing was taken into account, a vast number of vocoder voice frames were processed and their voice quality was assessed by using the perceptual evaluation of speech quality (PESQ) method. PESQ is an objective method for speech quality assessment of narrow-band telephone networks and speech codecs developed by the International Telecommunication Union (ITU) [15] [16]. Results of the PESQ tests showed that the stolen bit has very low impact on voice quality and intelligibility hence, its effect is negligible. PESQ test results for randomly selected synthesized speech files are shown in Table 1.

TABLE 1. MEAN OPINION SCORES OF RANDOMLY SELECTED AND SYNTHESIZED SPEECH FILES AFTER PROPOSED BIT STEALING PROCEDURE APPLIED

	Raw MOS	MOS LQO
File 1	4.392	4.480
File 2	4.416	4.496
File 3	4.434	4.508
File 4	4.353	4.453
File 5	4.419	4.498

After bit stealing, 48 bit long vocoder voice frame became suitable for 2/3 rate PCC. The proposed PCC can obtain a 1/2 rate convolutional code with constraint length 12 and the generator polynomial [6765 4627] [17]. The block diagrams of the proposed FEC encoder and decoder are given in Figure 6 and Figure 7, respectively.

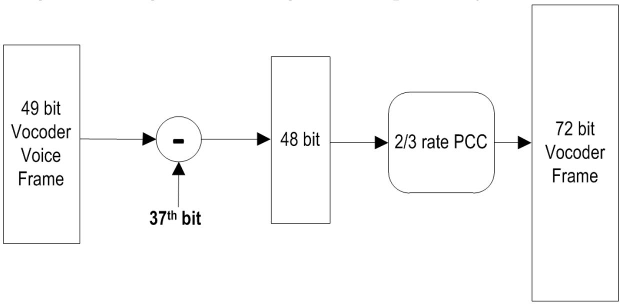

Figure 6. Block diagram of proposed 2/3 convolutional FEC encoder for vocoder.

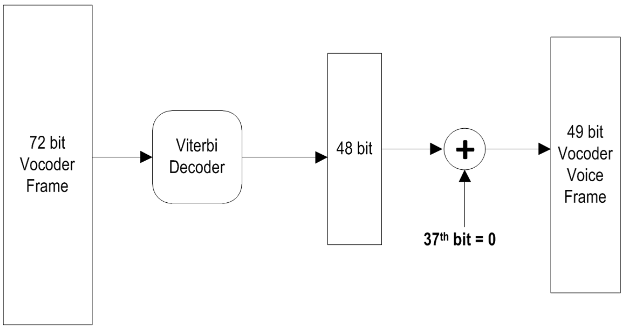

Figure 7. Block diagram of proposed 2/3 convolutional FEC decoder for vocoder.

IV. Performance Analysis

The bit error probabilities of the proposed FEC scheme are evaluated for four-level frequency-shift keying (4-FSK) modulation by Monte Carlo simulations. In the simulations, additive white Gaussian noise (AWGN) channel is employed. For comparison, the BER curves of uncoded 4-FSK, AMBE standard, Golay codes, hybrid Golay and 5/6 rate PCC FEC and the proposed 2/3 rate PCC FEC are also included in Figure 8.

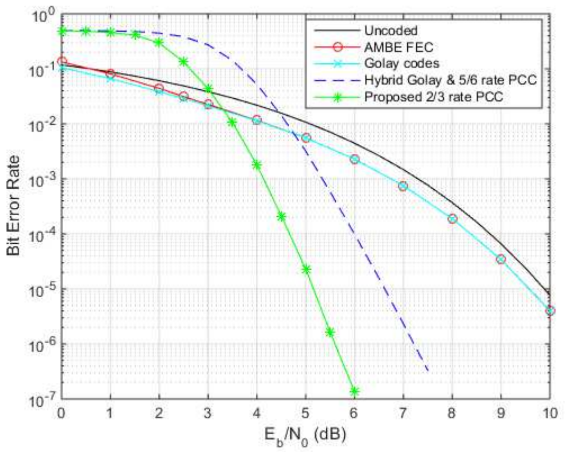

Figure 8. BER performance of uncoded, AMBE FEC, Golay (23,12) codes (AMBE FEC without scrambler), hybrid Golay (23,12) and 5/6 rate PCC and proposed 2/3 rate PCC.

Compared to uncoded 4-FSK, the AMBE FEC scheme provides an E_b/N_0 advantage of approximately 0.31 dB for a BER value of P_b=10^{-5}. The hybrid Golay and 5/6 rate PCC FEC scheme [8] provides an E_b/N_0 advantage of approximately 3.35 dB with respect to the AMBE FEC scheme. Relative to the hybrid Golay and 5/6 rate PCC FEC scheme, the proposed 2/3 rate punctured convolutional code FEC scheme provides an E_b/N_0 advantage of approximately 1.06 dB. Moreover, 2/3 rate punctured convolutional code FEC achieves 4.41 dB coding gain with respect to the AMBE FEC. Also, it can be clearly seen that the Golay code without data dependent scrambler (cyan curve), which is performance of Golay (24,12) and Golay (23,12) codes in sequence, is better than the AMBE FEC due to the high number of bit errors caused by the scrambler in low E_b/N_0 values.

V. Conclusion

For any of the digital radio systems listed in the introduction section, the received vocoder frames are conveyed to DVSI's AMBE 3000 vocoder IC or vocoder software library within the radio central processing unit or digital signal processing without any interpretation or processing. Since voice frames are protected only by vocoder FEC, it is easy to change vocoder FEC to obtain more coding gain without modifying the air interface protocols used in radios. In this work, bit stealing enabled us to use 2/3 rate PCC inside the AMBE 3600 bps vocoder while preserving audio quality, as shown in the PESQ test results. Audio quality measurements are given in terms of mean opinion score and obtained using PESQ method. With the help of 2/3 rate PCC, we obtained 1.51 dB and 4.41 dB coding gain compared to the study in [8] and AMBE 3600 bps FEC respectively. Increased coding gain provides increased voice quality and procures higher communication link quality in noisy environments. Increased coding gain may also help to extend battery consumption in battery powered radio. By utilizing the proposed FEC, we need less transmission power to achieve the same performance and communication range than that available with the present FEC.

VI. Future Work

For further improvement on AMBE FEC performance, unequal error protection techniques can be utilized to obtain more coding gain or increased voice quality. For future work, unequal error protection schemes will be evaluated and applied in order to enhance voice quality or coding gain.

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